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(54) Title: LOUDSPEAKER ARRAY FOR ENLARGED SWEET SPOT

(57) Abstract

The invention is a method of creating an enlarged listening sweet spot for multiloudspeaker audio reproduction. The method employs a plurality of audio drivers (52, 56) displaced generally in a horizontal dimension for a vertically oriented head of a listener (60), the drivers operating over a plurality of different passbands (42, 46). Higher frequency drivers (55, 59) are located closer to one another and displaced more towards a center line of the listening space than lower frequency drivers (53, 57), thereby causing a smaller change in acoustic ear signals for listeners (60) seated away from the designed–for listening position, but without causing increased low frequency signal capacity requirements for phantom images generally outside the array (50). Specially adapted signal processing accounting for the layout of the various drivers and their associated crossover networks (40) is employed to create desired audio images.

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LOUDSPEAKER ARRAY FOR ENLARGED SWEET SPOT

Field of the Invention

This invention relates generally to the fields of
5   audio signal reproduction and audio signal processing,
and more particularly to a system for increasing the
area over which a satisfactory audio illusion is created
and maintained, relative to prior art audio reproduction
systems. The method may employ a multi-way loudspeaker
10   pair with drivers operating over diverse frequency
ranges arrayed generally in a horizontal dimension (for
a normally-oriented head of a listener) and with
higher-frequency drivers generally closer together and
displaced more towards the center of the listening space
15   than lower-frequency drivers, and specially-adapted
signal processing components for audio imaging to create
or maintain desirable audio imaging.

Background of the Invention

20   The history of stereophonic sound ("stereophony"
or, more commonly and colloquially, "stereo") includes a
number of methods of recording sounds and another number
of methods for playing those recorded signals back to a
listener or listeners. While it has always been an
25   accepted idea that a listener should be "transported" to
another acoustical space, such as the acoustic space
occupied by an audience member at a live concert or a
more synthetic, more conceptual, space for many modern
popular recordings in which there was no actual
30   performance in front of a live audience, the methods
used for this "transporting" have largely failed in that

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goal. A reason for the failure has been that no
systematic, rigorous method was usually applied in
designing the various systems, designers and recording
personnel frequently instead relying primarily upon
5   largely unscientific principles and serendipity to
achieve their goals. That billions of commercial
recordings have been sold and broadcast is more a
statement of the appeal of the content of the recordings
than the ability to transport the listener into another
10   space. However, workers such as Schroeder and Atal, and
Cooper and Bauck, have devised playback systems which
employ signal processing methods which are firmly footed
in engineering science and based on the concept that
particular signals will be placed in and around the ears
15   of one or more listeners so that it becomes the task of
the producer of the program material to provide a
version of the "desired" signals.

These latter-day methods, though currently far in
the minority of systems and recordings purchased by
20   consumers to date, can be extraordinarily effective in
transporting the listener to another, believable,
acoustic space, when properly designed. Perhaps an
indication of the failure of traditional systems to
perform as hoped, and of the success of the latter-day
25   systems, is that the newer systems are often called "3D
audio," "3D sound," and the like ("3D" meaning
three-dimensional), and the vernacular use of
"stereophonic" often refers to the earlier systems. A
simple translation of "stereophonic" from its Greek-root
30   components means "of or relating to three-dimensional
sound." Thus, with the advent of practical
implementations of the latter-day systems, the audio

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community found it necessary to coin a new phrase, thus
"3D audio" and the like.

In keeping with current usage, we will use the
current term, 3D audio, to refer to the latter-day
5   systems. These systems typically employ some kind of
circuitry or algorithm which compensates for the fact
that sound emanating from each of two loudspeakers
impinges on both ears of a listener, so that, for
example, sound radiating from a left-placed loudspeaker
10   of a pair of loudspeakers travels to the left ear of a
listener, but also travels to the right ear of a
listener, this latter sound being called crosstalk. The
transmission from each loudspeaker to each ear can be
anticipated by designing the circuitry or algorithm,
15   from knowledge of so-called head-related transfer
functions (HRTFs), so that when the circuit or
algorithm, taken together with at least two
loudspeakers, all as a unit, can separately and
distinctly control the sounds at the ears of one or more
20   listeners. It is also possible to correct for frequency
response aberrations caused by the diffraction of the
listener's head so that a natural timbre is perceived by
the listener.

It is known in the art, especially in the patents
25   of Cooper and Bauck, that improved performance can be
achieved by deliberately modifying the filters
comprising the crosstalk cancelling circuitry or
algorithms or related circuitry or algorithms from their
strict specifications from HRTFs. For example, it may
30   be necessary in some cases to modify the filters in such
a way as to make them stable or otherwise realizable.
Other modifications are known in the art, such as using
HRTFs measured from a model mannequin head rather than

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the listener's own head, the use of minimum phase
transfer functions, the use of simplified head models
such as smoothed HRTFs, spheres, or two points in free
space (for ears), and the use of delays to convert

5   noncausal filters into causal filters. Some deviations
from the full HRTF specifications may be quite extreme,
for instance, following the HRTF specification up to
only some 600 Hz and allowing factors other than the
most precise imaging to specify the response above 600

10 Hz. Any such modification, while deviating from the
strict specification of the listener's own HRTFs, may be
considered to be advantageous, either for the sake of
performance or economies or both. Also, such
modifications may result in less than perfect

15 cancellation of crosstalk and/or less than perfect
correction of timbre. Nonetheless, we will refer to all
such devices as crosstalk cancellers. Crosstalk
cancellers are the heart of most 3D audio systems,
allowing predetermined control of signals at the ears of

20 the listener or listeners, thus removing many elements
of luck from the playback experience. It is therefore
an object of the invention that any crosstalk canceller
with any of the several described modifications or other
modifications may be used either explicitly or

25 implicitly as the imaging component of the invention.

One application of crosstalk cancellation is in
playing back recordings made with an acoustical
mannequin, a dummy head with microphones placed in its
ear canals or thereabouts. Such a recording-playback

30 system results in the most realistic impression of being
transported to another space.

Another application of crosstalk cancellers is as
part of an imaging circuit or algorithm, a so-called

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speaker-spreader or layout reformatter such as described
by Schroeder and Atal, and Cooper and Bauck. In this
application, the listener can receive the impression
that, for example, a pair of loudspeakers which is
5   placed on the sides of a television receiver cabinet,
much too close for perceiving any readily noticeable
amount of stage width, appear to be farther apart, with
well-defined sounds apparently emanating from points in
space where there are no actual loudspeakers, a "virtual
10   loudspeaker" impression. In this application, it is
most common for the input signals to be any kind of
ordinary stereo; the input signals may also be provided
by a home theater or multichannel television audio
decoder, providing five or more channels of audio
15   signals.

    Still another application of the principle of
crosstalk cancellation is in the creation of
interactively-controlled sound sources (and their
reflections in an acoustic environment, if desired) such
20   as would exist in computer-based or game-console-based
games, when the sounds for those games are presented to
the player or players over loudspeakers.

    So it is seen that a crosstalk canceller is a basic
component of controlling signals at the ears of a
25   listener, usable with either binaurally recorded
programs or with any kind of traditional stereo
programs, for the general enhancement thereof.

    Playback systems which do not effectively use a
crosstalk canceller are also sometimes known as 3D.
30   Such systems can create the impression that sound is
arriving from points in space where there are no actual
loudspeakers, but rather than provide the impression
that there are virtual loudspeakers or other spatially

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discrete or distinct sources, the impression is that of
a wall of sound with little or no impression of
spatially discrete sources. To the extent that these
systems benefit from placing loudspeakers close together

5   (as described below), they may also benefit from the
invention. And of course, enhancement of these
"nondiscrete" systems is possible by the use of the
virtual loudspeaker concept.

        It is an aspect of the invention that it may be

10   used or combined with any type of 3D imaging circuit or
algorithm, whether "discrete 3D" or "wall-of-sound 3D."

        With the general framework established, we may now
begin to discuss a specific problem that exists in
essentially all prior-art audio systems, whether of the

15   traditional or 3D variety. Essentially all such systems
have a listening area in which the sound impression is
best. Listeners in that area receive an impression that
is better than at any other place in the playback room
or listening space. Typically, there are two

20   loudspeakers and the favored area is on a line bisecting
a line segment drawn between the two loudspeakers, and
more particularly at a specified distance or other
geometrical relationship to the loudspeakers. Wherever
the favored region is, it is commonly called the "sweet

25   spot," and we will use that terminology here, even
though "spot" may tend to imply "point" rather than
"region." The sweet spot is restricted in its extent,
frequently being so small that only one person can enjoy
the best spatial impression at one time, whether for

30   traditional or 3D stereo; the sweet spot size is
sometimes so small that even a single listener may feel
constrained as to where he or she should hold his or her
head to fully enjoy the sweet spot. Usually the sweet

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spot is an elongated region, really rather prolate
ellipsoidal in shape, allowing listeners to move in and
out along the bisecting line, or up and down while
remaining mostly in the bisecting plane, but being very
5   unforgiving with respect to listener movement to the
left and right, over wide variations in a standard
two-loudspeaker setup. This is the most unfortunate
direction in which to have a small extent of the sweet
spot, since it is most commonly desired that multiple
10   listeners be seated abreast of one another and not lined
up nose-to-nape.

With the advent of practical 3D audio systems and
the associated ability to precisely control the sounds
at the listener's ears, it is common for listeners to
15   perceive that the sweet spot is smaller than they are
accustomed to with prior experience listening to
ordinary stereo systems. It has been conjectured by
Bauck and Cooper (such conjecture borne out informally
by the experience of many listeners to such 3D systems),
20   that the sweet spot is not actually smaller, but, since
it is much sweeter, listeners tend to feel more deprived
upon moving out of the sweet spot. Also, the rate of
deprivation with respect to movement away from the
optimum position would appear to be greater, perhaps
25   lending even more feeling that the sweet spot is rather
small.

Regardless of the nature of the playback system
(traditional or 3D), it is always desirable to make the
sweet spot larger. It is, therefore, an object of this
30   invention to do so.

One reason that there is a sweet spot is that with
reproduction with two or more loudspeakers, the signals
at the listener's ears are formed by the interference

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(summation) of acoustic waves emanating from the
loudspeakers. With two loudspeakers, the field can be
controlled precisely (assuming the absence of resonant
structures) at only two points. Presumably, those
5   points are to be at the listener's ears. Whether the
ear signals are a result of a so-called 3D system or any
other technique, if the listener moves his or her head
so that the ears are no longer at the designated
positions, image distortion will appear, caused by
10   unintended ear signals created by unanticipated
interference. The primary causes of the changing
interference are differing times-of-arrival due to
differing loudspeaker-to-listener distances, followed in
importance by amplitude variations of the impinging
15   waves due to the same varying distances (aggravated by
the listener sitting close to the loudspeakers), and
reflections from any uncompensated reflections (improved
by the listener sitting close to the loudspeakers).

     An important aspect of this diffraction problem is
20   that for a given amount of movement of the listener's
head from the designed-for position it is wavelength
dependent. Ear signals at higher frequencies are
affected relatively more than those at lower frequencies
because the given amount of movement is a larger
25   fraction of a wavelength (or larger number of
wavelengths) at the higher frequencies.

     In prior art systems, the effects of a listener
moving out of the sweet spot are well-known, even by
casual listeners. For example, a vocalist who initially
30   appears as a centered phantom image midway between two
loudspeakers when the listener is on the bisecting line
then appears to subsequently shift towards the nearer
loudspeaker when the listener moves away from the

bisecting line. The effect is so pronounced that the sound image collapse into the nearer loudspeaker is nearly complete when the listener's head is only a few inches closer to the one loudspeaker than the other. This is the well-known precedence effect, sometimes called the Haas effect after one of its early researchers. It is usually thought to be a psychoacoustic effect, perhaps with its origins in the processing of the inner ear or brain. If that is the case, it may be an evolutionary adaptation to allow accurate localization of sounds in reflective environments. However, it is possible that the effect is also rooted in physical acoustics, a hypothesis that has not been fully investigated. In any event, the amount of image shift as a function of time-of-arrival differences from two sources has been studied thoroughly, with the result that the farther the listener is from the bisecting line, the farther the perceived shift of the center phantom image. It should be noted that the perceived image distortion due to this effect is not, strictly speaking, a shift, but is accompanied by an increase in the spatial extent of the image, or, more oddly, a kind of ambiguity or uncertainty as to the actual location of the image.

One prior art method attempts to reduce the shifting of phantom images by the use of specially designed loudspeakers. Researchers investigating the precedence effect found that the shift of a previously centered phantom image could be partially compensated by increasing the level of the later-arriving sound, that is, by increasing the signal gain of the more distant loudspeaker of the pair. In fact, experimentally derived plots have been published which show how much

the gain has to be increased, as a function of
time-of-arrival differences, to bring the image back to
the center, or approximately so. Such compensations,
though not precise and not resulting in a well-formed
re-centered phantom, have been found useful enough by a
few loudspeaker manufacturers that they have made
loudspeakers which had radiation patterns so that as a
listener moves from the bisecting line, he or she moves
more directly into the main lobe of the more distant
loudspeaker. A version of this plan has the listener
orienting his or her conventionally-designed
loudspeakers so that their main radiation lobes cross in
front of the specified listening position (over tow-in).
Some found either technique to be helpful, but the
compensation is only approximate, and less effective at
low frequencies due to the relative impossibility of
creating directional radiation patterns at those
frequencies. Nonetheless, it is an object of this
invention that this type of radiation control may be
combined with the novel techniques described herein to
accommodate more types of solutions to the sweet spot
problem.

    Another prior art technique, introduced by Cooper
and Bauck, used a method (which is independent of the
present invention) of alleviating the perceived sweet
spot problem in 3D systems by modifying the responses of
the acoustically-specified imaging filters at the higher
frequencies, effectively allowing gradual transition to
"default" imaging of the affected frequencies at the
loudspeakers. Listeners seem to prefer having the
higher frequencies remain mostly stationary with head
movements than to have them flitting around or be
otherwise poorly imaged. Indeed, the sweet spot can in

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fact be enlarged by modifying the filters down to lower
frequencies, but at the expense of more and more of the
higher frequencies falling into the loudspeakers, a
trade-off in sweet spot size for "sweetness." It is an
5   object of the invention that it may be combined with
such prior art methods.

A crucial observation is that the time-of-arrival
differences from two loudspeakers to either ear of a
listener, as he or she moves about on either side of the
10   bisecting line, is diminished if the loudspeakers are
close together. A simple plot of time-of-arrival
differences is shown in FIG. 1, for a single point in
space. The hyperbolic curves represent contours of
equal time-of-arrival differences, in milliseconds. The
15   horizontal and vertical axes are positions of the point
in space, in meters. The small, heavy circles represent
the locations of the two loudspeakers, modeled as point
sources. A is calculated for loudspeakers at a distance
of 1.5 meters, while B is calculated for a loudspeaker
20   distance of 0.5 meters. (For convenience, the
loudspeaker spacing and the line between loudspeakers
will be referred to as the baseline distance, or simply
the baseline.) It is apparent from these contour plots
that the short-baseline array results in smaller
25   time-of-arrival anomalies for the same amount of
displacement from the center line. While this simple
model and analysis does not include the effects of the
listener's HRTFs or indeed the fact that a
normally-endowed listener has two ears, it nevertheless
30   illustrates the basic principle. An analysis using a
more realistic model will be explored in detail shortly.

That fact that short-baseline arrays hold some
advantages was noticed some years ago by Cooper and

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Bauck. Other researchers have more recently studied the
advantages of this approach. The ultimate
short-baseline array is the monopole-dipole
("middle-side") array of Lauridsen, and its improvements
5   as taught by Cooper and Bauck. Of course, with the
technology of virtual loudspeakers, one may consider
deliberately creating a short-baseline array, then
expanding the apparent stage width with the appropriate
signal processing, for an expanded sweet spot, but at a
10   cost of trading stability of outlying images for
improved stability of near-center images, depending upon
the details of a particular design. In other cases, a
short-baseline array may be dictated by other needs,
such as the need to attach loudspeakers on the sides of
15   a television or computer video monitor, or the practical
difficulty of locating the several loudspeakers common
in current home theaters in their optimum locations.

It is nearly universal practice in loudspeaker
design to configure the tweeters and woofers of a
20   two-way loudspeaker, or more generally the various
transducing drive units (acoustical emitters) covering
different frequency bands in a multiway loudspeaker, in
a primarily vertical direction. While there are
exceptions, in which for example a midrange driver may
25   be located beside a tweeter, perhaps with one or both of
them comprising a "line source" or ribbon-style driver,
such side-by-side placement is usually accepted as a
compromise in the pursuit of other design goals, and it
is usually desired that those drivers should be as close
30   together as possible, horizontally, to maintain signal
integrity at the listeners' ears.

There have been attempts to create loudspeaker
arrays using horizontally-oriented multi-passband drive

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units. Electromagnetic versions of such arrays are also
used from time to time in communications and radar
antennas. In either application, the intent is to
control, at least partly, the radiation pattern at
5   various frequencies, usually with the intent that it
maintain a constant shape, or beamwidth, at all
frequencies of the intended range of operation. Such a
goal can be attained, at least partially, by creating an
array which is effectively the same length at all
10   frequencies, as measured in number of wavelengths at
each frequency. The normal procedure for doing this is
to progressively low pass filter the feed signals to the
elements of the array more severely for elements lying
more towards the ends of the array. This technique
15   remains largely an obscure curiosity in the field of
audio reproduction due to the enormous range of
frequencies normally encountered (some ten octaves for
high fidelity reproduction) and the fact that to attain
significant control of the radiation pattern over
20   important portions of the audible spectrum would require
arrays of such a large size as to render them
impractical. Some proposals have been more modest,
suggesting that beamwidth control over as little as an
octave can be effective for applications such as sound
25   reinforcement, but this is not an application in which
more than one loudspeaker is used to form effective
audio phantom images (the field addressed by the
invention) nor does the invention teach the formation of
constant radiation patterns over frequency variation,
30   although there may well be a tendency towards such
behavior as a side effect.

     Another prior art loudspeaker employing a
horizontal array is that of Polk. However, the drive

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units of this device are arrayed in this manner for
other purposes, do not employ imaging circuitry, do not
enlarge the sweet spot, and in other ways do not
anticipate the invention.

5   While the use of a short baseline alleviates the
sweet spot problem, another problem arises; the degree
of the problem depends on the location and frequency
content of a virtual image in a 3D system. Consider
that if a natural image containing large amounts of low
10   frequencies relative to other frequencies appears
towards a listener's left-hand side (90°
counterclockwise from above from the nose which is
considered to be at 0°), the dominant air particle
motion in the vicinity of the head is to and fro,
15   parallel to a line through the ears. In order for two
front-placed loudspeakers to recreate such a
low-frequency motion, they must operate with
substantially opposite polarity on similar signals.
This constitutes, at the lower frequencies, an
20   approximation to the well-known and much-studied
acoustic dipole. The problem with this arrangement is
that the two loudspeakers tend to cancel one another's
low frequency sound (i.e., relatively little low
frequency energy is radiated towards the listener).
25   Consequently, potentially large signals must be applied
to the loudspeakers, requiring large amplification
factors and large excursions of the loudspeaker
radiating surfaces, a practical problem in applications
of such 3D systems as virtual home theaters and video
30   games reproducing side-placed low frequency sound
effects, as well as in 3D processing and playback of
many recordings of ordinary music. Another disadvantage
is that large amounts of acoustic energy are reflected

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around the room before finally arriving at the listener,
introducing still more unaccounted-for factors into this
playback method

A closely related scenario is that of a binaural
5   recording being played over a crosstalk canceller.   In
this application, the low-frequency problem at first
appears to be even worse, since the filter specification
is for even more bass signal for a virtual source
towards the listener's left, as taught most clearly by
10   Cooper and Bauck in their explanation of
sum-and-difference style of signal processing.
Depending on the loudspeaker angle (as seen by the
listener), the bass response of the left-minus-right (L
- R) component, that which is predominant in the
15   placement of the left-oriented image, at first
inspection seems to be such as to make the whole
enterprise nearly impractical, showing a first-order
increasing slope (20 dB per decade of frequency) with
decreasing frequency, and with the onset of the slope
20   occurring at a higher frequency with more closely-spaced
loudspeakers.   However, it is important to realize that
a naturally-occurring image at 90° necessarily contains
relatively little L - R information in the low
frequencies, since the ear signals are nearly identical
25   in both amplitude and phase.   Therefore, although a
large L - R gain might be specified, the L - R signal is
small, so the filtered signal might still be of a
reasonable size, assuming that the loudspeakers are not
too close together. The only practical problem is
30   maintaining a good signal to noise ratio, but this is
generally not a problem with either analog or digital
implementations. The net result is that the extent of

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the problem is essentially the same as creating a
virtual source as described in the preceding paragraph.

More severe scenarios are easily imagined. It is
quite easy to conceive or create a stereo signal which
5   does not correspond to any natural sound image and which
will wreak havoc when played through, for example a
loudspeaker-spreader or other layout reformatter or
crosstalk canceller, all examples of 3D audio systems.
For example, a bass guitar in one originating channel of
10   a conventional stereo formatted signal, with silence in
the other channel, when played over a crosstalk
canceller, is highly unnatural; the playback system
attempts to place the sound of a bass guitar in one ear
of the listener and silence in the other ear, an
15   extremely demanding task at any reasonable playback
volume.

In any of the examples described above, the demands
on low-frequency signal excursions in both the
amplifiers and loudspeakers increase, that is, get
20   worse, the closer together the loudspeakers are placed.
Thus, the desirable effects of a short-baseline array
are offset by the greatly increased signal handling
capacity required to realized the necessary signals,
both electronic and acoustic.

25   This circumstance, that of increased low-frequency
signal capacity requirements with shorter-baseline
arrays, is extremely unfortunate, as it compounds with
another aspect of low-frequency reproduction of audio
signals. As is well known by audio engineers, in an
30   ordinary stereo set, (i.e., one not required to place
low-frequency images outside the spatial extent of the
usual two-loudspeaker layout or, for that matter, any
single-loudspeaker audio reproduction system), the

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adequate reproduction of the lower frequencies a priori
requires that much larger volumes of air be moved. This
is normally accomplished by using drivers of larger
diameter and with a much larger linear excursion

5   capability than needed for reproduction of higher
frequencies. Similarly, most of the linear signal
excursion range of the associated amplifiers is used up
by the lower frequencies, with the smaller higher
frequency components appearing to ride atop the more

10   slowly undulating low frequency components.

The problems of creating adequate signal levels for
low-frequency program material placed as virtual images
generally outside the extent of a two-loudspeaker array
are not merely hypothetical examples. The inventor has

15   observed precisely the behavior that he describes on
numerous occasions when demonstrating various types of
3D audio programs. In today's commercial environment,
with consumers demanding more realism from their audio
systems, with the popularity of home theaters and games

20   and the associated preponderance of high-level,
low-frequency, side-placed sound images of special
effects (frequently played over actual loudspeakers in a
full home theater setup), and the emergence of computers
with attached audio systems intended for playing games

25   and simulating home theater systems as "virtual
theaters," the scenarios described herein are very real
indeed. It would surprise many just how quickly even a
rugged, well-designed loudspeaker system can reach its
limits under such circumstances, not to mention the

30   inexpensively-made loudspeakers often associated with
computers.

Another problem compounds with short-baseline
arrays. To overcome the large low-frequency excursion

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requirements, one might decide to use larger woofers. However, if the woofers are round, it becomes rather nonsensical to find a way to place them close together. One may resort to oval or rectangular radiating surfaces, but these tend to have still other problems. Also, placing large loudspeakers close together may unacceptably compromise the practical and aesthetic design of a product such as a television or computer video monitor.

Summary

Briefly, according to one embodiment of the invention, an audio reproduction system is provided including means for providing any number of audio inputs, means for providing audio imaging using a crosstalk canceller, and a pair of two-way loudspeaker systems, each arrayed with woofer and tweeter substantially horizontally for normally-oriented heads of one or more listeners, such loudspeaker systems comprising frequency-selective crossover circuits to separate and route signals into a left woofer and tweeter pair and a right woofer and tweeter pair, the woofer and tweeter of each pair arranged so that the left and right tweeters are closer together than the left and right woofers, so that time-of-arrival differences from the tweeters vary less with off-center listeners than do time-of-arrival differences from the woofers, for similarly off-center listeners.

Brief Description of the Drawings

The invention, together with further objects and advantages thereof, may be understood by reference to

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the following description taken in conjunction with the
accompanying drawings.

　　　　FIG. 1A is a plot of equal time-of-arrival contours
for a simple model of a prior art loudspeaker array.

5　　　　　FIG. 1B is a plot of equal time-of-arrival contours
for a simple model of another prior art loudspeaker
array.

　　　　FIG. 2 is a generalized block diagram of a sound
reproduction system under an illustrated embodiment of

10　　　the invention.

　　　　FIG. 3A is a block diagram of an illustrated
example of the imaging circuit component of an
embodiment of the invention, shown in support of an
analysis of the invention.

15　　　　　FIG. 3B is a block diagram of another illustrated
example of the imaging circuit component of FIG. 2,
shown in support of an analysis of the invention.

　　　　FIG. 4 is a plan view layout diagram showing
examples of transfer functions of the system of FIG. 2

20　　　from a loudspeaker to a listener.

　　　　FIG. 5A is a plot of the magnitude of the transfer
function of an imaging filter of the imaging component
of FIG. 3B, for a prior art loudspeaker array comprising
full-range loudspeakers at ±20° relative to a listener.

25　　　　　FIG. 5B is a plot of the magnitude of the transfer
function of another imaging filter of the imaging
component of FIG. 3B, for a prior art loudspeaker array
comprising full-range loudspeakers at ±20° relative to a
listener.

30　　　　　FIG. 5C is a plot of the magnitude of the transfer
function of an imaging filter of the imaging component
of FIG. 3A, for a prior art loudspeaker array comprising
full-range loudspeakers at ±20° relative to a listener.

FIG. 5D is a plot of the magnitude of the transfer
function of another imaging filter of the imaging
component of FIG. 3A, for a prior art loudspeaker array
comprising full-range loudspeakers at ±20° relative to a

5   listener.

FIG. 6A is a plot of the magnitude of the transfer
function of an imaging filter of the imaging component
of FIG. 3B, for a prior art loudspeaker array comprising
full-range loudspeakers at ±3° relative to a listener.

10   FIG. 6B is a plot of the magnitude of the transfer
function of another imaging filter of the imaging
component of FIG. 3B, for a prior art loudspeaker array
comprising full-range loudspeakers at ±3° relative to a
listener.

15   FIG. 6C is a plot of the magnitude of the transfer
function of an imaging filter of the imaging component
of FIG. 3A, for a prior art loudspeaker array comprising
full-range loudspeakers at ±3° relative to a listener.

FIG. 6D is a plot of the magnitude of the transfer

20   function of another imaging filter of the imaging
component of FIG. 3A, for a prior art loudspeaker array
comprising full-range loudspeakers at ±3° relative to a
listener.

FIG. 7A is a plot of the magnitude of the transfer

25   function of an imaging filter of the imaging component
of FIG. 3B, for an embodiment of the invention, a
loudspeaker array comprising woofers at ±20° and
tweeters at ±3°, both relative to a listener.

FIG. 7B is a plot of the magnitude of the transfer

30   function of another imaging filter of the imaging
component of FIG. 3B, for an embodiment of the
invention, a loudspeaker array comprising woofers at
±20° and tweeters at ±3°, both relative to a listener.

FIG. 7C is a plot of the magnitude of the transfer function of an imaging filter of the imaging component of FIG. 3A, for an embodiment of the invention, a loudspeaker array comprising woofers at ±20° and

5   tweeters at ±3°, both relative to a listener.

FIG. 7D is a plot of the magnitude of the transfer function of another imaging filter of the imaging component of FIG. 3A, an embodiment of the invention, a loudspeaker array comprising woofers at ±20° and

10   tweeters at ±3°, both relative to a listener.

FIG. 8 is a plot of an error measure of an analysis of a prior art loudspeaker array comprising full-range loudspeakers at ±20°, relative to a listener, plotted versus listener displacement left and right, and forward

15   and backward, of an optimum seating position.

FIG. 9 is a plot of an error measure of an analysis of a prior art loudspeaker array comprising full-range loudspeakers at ±3°, relative to a listener, plotted versus listener displacement left and right, and forward

20   and backward, of an optimum seating position.

FIG. 10 is a plot of an error measure of an analysis of an embodiment of the invention, a loudspeaker array comprising woofers at ±20° and tweeters at ±3°, both relative to a listener, plotted

25   versus listener displacement left and right, and forward and backward, of an optimum seating position.

FIG. 11 is a television monitor or computer video monitor, according to the invention.

FIG. 12 is a wide-format television receiver

30   console, with a three-way loudspeaker array configured according to the invention for vertical dispersion control from the tweeters.

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Detailed Description of Preferred Embodiment

FIG. 2 is a generalized block diagram of a specific embodiment of an audio reproduction system 10 according to an illustrated embodiment of the invention. The

5   audio system 10 provides means 20 for coupling one or more audio signals into audio processing circuitry (e.g., computer algorithm) 30. Processed audio signals are then coupled into loudspeaker frequency selective crossover networks 40, including a separate crossover

10   network 42 for the left loudspeaker and a separate crossover network 46 for the right loudspeaker. The left crossover 42 may be comprised of a low pass filter 43 to separate and route low frequency signals to left woofer 53, and a high pass filter 45 to separate and

15   route high frequency signals to left tweeter 55. Similarly, right crossover 46 may be comprised of a low pass filter 47 to separate and route low frequency signals to right woofer 57, and a high pass filter 49 to separate and route high frequency signals to right

20   tweeter 59. The collection of left loudspeaker 52 and right loudspeaker 56 are referred to as the loudspeaker array 50. In this embodiment, two-way left and right loudspeaker systems are described, being the simplest embodiment of the invention, but it will be understood

25   that three-way, or generally, multiway, loudspeakers may be accommodated, and the loudspeaker systems may be arrayed in other ways and in other numbers than the usual symmetric left-right pair, according to the invention. Left loudspeaker 52 is comprised of a left

30   woofer 53 and a left tweeter 55, and the right loudspeaker 56 is comprised of a right woofer 57 and a right tweeter 59. The acoustically radiating elements, woofers 53 and 57 and tweeters 55 and 59, are arranged

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into a substantially horizontal array, for a
normally-oriented listener's head. (If a listener's
head 60 is not oriented in a normal, upright, position,
then the array 50 may be reoriented so as to maintain
5   the same approximate geometrical relationship to the
head of the listener, or the imaging circuitry 30 may be
adapted accordingly. However, an extremely precise
upright alignment of the listener's head 60 is not
normally required.)
10   It is also anticipated that the left loudspeaker 52
and the right loudspeaker 56, instead of being comprised
of separate woofers and tweeters, may alternately or
additionally comprise a so-called Walsh type driver, a
roughly megaphone-shaped truncated cone from which the
15   higher frequencies radiate preferentially from the
larger-diameter regions; however, according to the
invention, instead of the cones being oriented
vertically with the small ends on top and the large ends
on the bottom as in normal usage, they are instead
20   oriented with the small end closer to the geometric
center line and the large ends being relatively farther
away from the centerline.
According to the invention, the tweeters 55 and 59
are to be substantially closer to the median line
25   connecting the center of the head of the listener 60 and
the midpoint of the loudspeaker array baseline than are
the woofers 53 and 57. In another embodiment, the
positions of the left tweeter 55 and the right tweeter
59 are interchanged, but not their electrical
30   connections, so that, for example, as the listener moves
to the right, off of the center line, he or she moves
more directly in line with the left-channel tweeter,
thereby providing a partial and automatic compensation

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of time-of-arrival error by amplitude adjustment means,
according to the teachings of the precedence effect, and
also providing additional image stability by geometrical
means. Also according to the invention, the imaging

5   circuitry or algorithm 30 is adapted to account for the
spatial layout of the various acoustical radiating
elements. These and other aspects of the invention,
including an improvement in size of sweet spot without
increased low frequency signal handling requirements

10   which characterize prior art loudspeaker arrays, will be
made clear by a simulation calculation of a specific
embodiment, below.

The operation of the embodiment of FIG. 2 will now
be described. Assume as an example that a virtual image

15   or virtual loudspeaker is to placed at 90°. The
tweeters 55, 59 are close together; with their high pass
crossovers 45, 49, they receive little low frequency
signal energy and thus are not subject to the
large-excursion signals that would otherwise be required

20   of loudspeakers at that close spacing operating in the
bass region. High frequency characteristics of
head-related transfer functions from the tweeter angles
are generally such as to provide enough transfer
function differences between a particular tweeter and

25   both ears (or both tweeters and one ear) as to provide
an acceptable solution to the required equations. Thus,
at the higher frequencies where the sweet spot for
wider-spaced loudspeakers would be the smallest, the
close tweeters provide an enlarged sweet spot.

30   The woofers, spaced farther apart than the
tweeters, receive little high frequency energy due to
their low pass crossover filters. Thus, they do not
impose the small sweet spot at high frequencies that

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they would if they received high frequency signals.   On
the other hand, being spaced farther apart than the
tweeters, they do not have the large signal excursion
requirements that they would if placed closer, say at
the tweeter positions.

The other significant component in the embodiment
of FIG. 2, the 3D imaging system 30, is designed to
account for the different spacings of the woofers and
tweeters, and, just as importantly, to account for their
crossover filters. Of course, conceptually, the part of
the 3D imaging system that needs to be designed
specifically for the loudspeaker array-crossover
combination is a crosstalk canceller, whether it appears
as a separate component or sequence of software
instructions, or whether is combined explicitly or
implicitly with other 3D imaging components or software
such as HRTF simulations in a complete virtual imaging
system.

In nearly all conventional, prior art loudspeaker
systems, the crossover networks are designed so that,
for example, in a two-way system, the woofer and tweeter
operate over substantially different passbands. A
typical configuration is to operate a woofer up to
around 2.5 KHz, where its operation is limited by a low
pass filter at frequencies above 2.5 KHz, and to operate
a tweeter at frequencies from 2.5 KHz upward, with its
operation limited at frequencies below 2.5 KHz by a high
pass filter. While the crossover filters do not have
abrupt transition bands (it is physically impossible),
there tends to be a small range of frequencies near the
crossover frequency which are radiated by both woofer
and tweeter, even though the woofer and tweeter operate
over substantially different passbands. Some designers

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will even go to great lengths to restrict that band of
commonality by specifying expensive, rapid-cutoff
crossover filters. However, there are examples of prior
art loudspeakers with two drivers operating over

5   different passbands, but which passbands have
considerable overlap. A typical example of this concept
has two lower-frequency drivers, possibly of identical
designs (diameter, electroacoustic parameters, etc.),
but with one of them limited by filters to a rather

10   narrower band of frequencies. For example, one driver,
commonly referred to as a woofer, may operate from the
lowest frequencies, say 40 Hz, to some 100 Hz, while the
other driver, perhaps referred to as a "mid-bass"
driver, is allowed to handle frequencies up to 2.5 KHz,

15   where a crossover passes the higher frequencies to a
tweeter. Such a design allows for increased
signal-handling capacity for the lowest frequencies
where it is needed most, but does not impose undesirable
directional effects at emitted frequencies from 100 Hz

20   to 2.5 KHz caused by the dual drivers operating over
somewhat shorter wavelengths where lobing could occur.
So, while there are prior art loudspeakers which have
drivers operating over substantially overlapping
frequency ranges, they are vertically-arrayed, do not

25   use the overlapping passbands to affect desirable
imaging effects, and do not operate with
specially-designed imaging circuitry or algorithms. It
is an object of the invention that horizontally-arrayed
drivers according to the invention may have

30   substantially overlapping passbands and such drivers are
to be accompanied by suitably modified imaging circuits,
as taught in the following discussion describing a
simulation of an embodiment of the invention.

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A simulation of a specific illustrated embodiment
of FIG. 2 will help to described the invention and to
clarify its teachings.

For a source of HRTFs, a spherical head model is
5   used (i.e., the head 60 is assumed to be a rigid sphere
0.18 meters in diameter with "ears" 62 and 63 designated
as points on a horizontal great circle displaced ±100°
from the "nose" 61 which is defined to be the point at
0°).   The sphere is assumed to be in a plane wave field
10   when a single source is present. Positive angles are
measured as counterclockwise rotations from the nose 61.
This model is used because it is convenient to acquire
(compute) HRTFs from any angle and is more than adequate
to describe and define the invention and its teachings.

15   The loudspeaker-listener layout example which is
examined here is typical of that experienced by a
personal computer user. In assumed free-field
conditions, the relevant parameters are:

• Distance from center of the head 60 to the plane
20   of the video monitor and loudspeaker array 50 is 0.6 m;

• Angular displacement of tweeters of the
invention, 55, 59, as measured from the center of the
head is ±3°;

• Angular displacement of woofers of the invention,
25   53, 57 as measured from the center of the head is ±20°;

• Angular displacement of prior art full-range
loudspeakers used for comparison is ±3°.

• Angular displacement of other prior art
full-range loudspeakers used for comparison is ±20°.

30   Although an assumption of a plane wave in the
vicinity of the head 60 and the statement that the
loudspeakers are only about 0.6 meters distant are,
strictly speaking, inconsistent (the loudspeakers would

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examined, as alluded to in the above geometric
description, are:

• Full-range prior art loudspeakers at
"conventional" video monitor spacing of ±20°;

5   • Full-range prior art loudspeakers at "very
close" spacing of ±3°;

• Modified short-baseline array according to the
invention with woofers 53 and 57 at ±20° and tweeters 55
and 59 at ±3°.

10   The various filter magnitude responses are plotted
for two types of symmetric topologies for convenience:
the lattice in FIG. 3A and the extended shuffler of
Cooper and Bauck in FIG. 3B. The phase responses, while
as important to proper image formation as the magnitude

15   responses, are not displayed for the present purposes of
examining low frequency signal capacity (magnitude)
requirements of various layouts, although they are used
in the simulation calculations; when a symbol is used to
refer to a transfer function, it is assumed to be a

20   complex-valued variable, as is normal for such analyses.

The lattice 70 of FIG. 3A may be used to implement
the imaging circuit or algorithm 30 of FIG. 2, as long
as there are no more than two inputs 20, as may the
shuffler 80 of FIG. 3B, as is known in the art; other

25   topologies are also possible, possibly employing
additional lattices or shufflers. The lattice 70
comprises the four filters of two unique transfer
functions (assuming geometrical layout symmetry, as in
the simulation), S' and A'. The shuffler 80 requires

30   only two filters with transfer functions Σ and Δ. The
lattice requires summing junctions 75 and 76. The
shuffler requires summing junctions 81, 82, 85, 86, with

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signal sign inversions at two of the inputs of summing
junctions 82 and 86, as indicated in FIG. 3B.

FIG. 4 serves to establish the naming convention
for acoustic transfer functions, generically, S and A.
Subscript α indicates a transfer function associated
with an "actual" loudspeaker, and subscript φ indicates
a transfer function of a "phantom," in this case the
desired image at 90°. With this notation, the imaging
filter transfer functions when using
symmetrically-placed full-range loudspeakers are

$$\Sigma = \frac{1}{2} \frac{S_\phi + A_\phi}{S_\alpha + A_\alpha}$$

$$\Delta = \frac{1}{2} \frac{S_\phi - A_\phi}{S_\alpha - A_\alpha}$$

$$S' = \Sigma + \Delta$$
$$A' = \Sigma - \Delta$$

In the simulation, the grid over which the
spherical-model head is moved for computing an error
measure is a square of 0.322 meters per side centered on
an x-y coordinate system which has as its origin the
center of the nominally-placed spherical head. For 50
frequency points f_n, n = {1, 2, . . ., 50), evenly
spaced between 20 Hz and 8000 Hz, frequency domain
signals for the left ear $E_L(f_n, x, y)$ and right ear

$E_R(f_n, x, y)$, and nominal-position ear signals $E_L(f_n, 0, 0)$ and $E_R(f_n, 0, 0)$, an error measure

$$\varepsilon(x, y) = 20 \log_{10}(\sum_{n=1}^{50} \left(|E_L(f_n, x, y)| - |E_L(f_n, 0, 0)|\right)^2 + \left(|E_R(f_n, x, y)| - |E_R(f_n, 0, 0)|\right)^2)$$

is computed. The choice of 8000 Hz as the highest
frequency to include in the error is a compromise
between including most of the significant audio range
for localization and allowing too many sphere-model
idiosyncracies to contribute. This is obviously a
signal-based error function--its precise connection to
perceived error is unknown. The x-y grid spacing was
1/4 of the shortest wavelength included in the above
sum, and there were 31 grid points in each direction,
resulting in the square of 0.322 m. (A grid spacing of
½ the shortest wavelength would be enough for adequate
spatial sampling, but the finer grid gives a nicer
surface plot.)

        For full-range prior art loudspeakers at the ±20°
positions (and a virtual image at 90°), the Σ, Δ, S′,
and A′ are shown in FIG. 5. Of particular importance
are the low-frequency levels of, for instance, the Σ and
Δ filters. The Σ filter, due to the factor of ½ in its
definition, levels off at low frequencies at -6 dB,
while the Δ response is some 9.3 dB higher, at 3.3 dB, a
quite tolerable level difference for most applications.
        The responses corresponding to full-range
loudspeakers at ±3° is shown in FIG. 6. Again, the Σ
response at low frequencies is at -6 dB, but the Δ

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response has increased drastically to 25.6 dB above the
Σ level.

For the modified array 50, the definition of the
acoustic transfer functions need to be modified to
account for the different locations of the drivers, as
well as for the presence of their respective crossover
filters. Let the transfer function of each woofer
crossover filter 43, 47 be C_w and the transfer function
of each tweeter crossover filter 45, 49 be C_t. Also,
let the generic transfer functions S and A of FIG. 4 be
specialized for the woofer and tweeter positions as S_w,
A_w, S_t, and A_t. With the help of FIG. 2, the new,
composite, "actual loudspeaker" acoustic transfer
functions can be seen to become

$$S_\alpha = C_W S_W + C_t S_t$$

and

$$A_\alpha = C_W A_W + C_t A_t.$$

These expressions can then be used in the calculation of
the imaging filters Σ, Δ, S′, and A′ in FIG. 3. The
crossover filters are simply modeled as first-order low
pass responses for C_w 43, 47 and first-order high pass
responses for C_t 45, 49, both having magnitude responses
which are -3 dB from their asymptotic values at 1 KHz.

For the simulation, no attempt is made to optimize
either the layout geometry or the crossover filter
response shapes. The main reason for picking 1 KHz for

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the crossover frequency is that the magnitudes of Δ, S',
and A' in FIG. 6 become large at around that frequency.
Also, practicalities such as driver characteristics
played little role in crossover selection.

5   The imaging filter responses for the modified array
50 of the invention are shown in FIG. 7. The low
frequency parts of these curves indeed resemble the low
frequency parts of the curves for ±20° full-range
loudspeakers in FIG. 5. Also, the high-frequency parts

10   of the responses of FIG. 7 resemble the high-frequency
parts of the responses in FIG. 6 for the very closely
spaced full-range loudspeakers at ±3°.

In FIG. 8, FIG. 9, and FIG. 10 are shown the error
function defined above for the three layouts considered,

15   over the x and y ranges mentioned above. In particular,
FIG. 8 shows the error which results from full-range
loudspeakers at ±20°, FIG. 9 shows the error from
full-range loudspeakers at ±3°, and FIG. 10 shows the
error from the modified array 50. All are plotted over

20   the same vertical range of 40 dB, from 7 dB to 47 dB.
Despite the lack of a firm connection to a
psychoacoustic error metric, these surface plots seem to
capture the subjective nature of the sweet spot. By
design, all have zero error ($-\infty$ dB) at $x = 0$, $y = 0$.

25   The implications of these figures are clear. The
±20° full-range array has a small sweet spot which is
longer in the y-direction than it is wide in the x-
direction, a phenomenon readily noticed by listeners of
both 3D and more standard stereo systems. The sweet

30   spot for ±3° full-range loudspeakers is much wider by
comparison. It is lower than the error plot for ±20°
loudspeakers at most points, with the exception at large
positive values of y and $|x|$. The error function for

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the modified array 50, FIG. 10, closely resembles that of the ±3° array, and in some regions is even slightly lower.

From the results shown from the simulation, it is seen that the modified short-baseline array 50 of the invention, with appropriately designed imaging circuitry, has very nearly the sweet spot size of a very short-baseline prior art array but the signal excursion requirements of a longer-baseline prior art array—it has the best characteristics of both.

The invention also has the advantage of having nearly zero incremental cost to implement, over conventional loudspeaker array-imaging circuit combinations, requiring only a new layout for the various drivers of the loudspeakers and a properly adapted imaging circuit. Neither of these changes comprises a significant recurring manufacturing cost; indeed, some expense may be saved if lower-power amplifiers or less expensive woofers can be used.

The applications for which the modified short-baseline array appears to be well-suited include that shown in FIG. 11, a computer or television video monitor which contains the modified array with woofers 103, 107 and tweeters 105, 109 and the associated imaging circuits (not displayed in the figure) such as required for virtual home theaters and game play. FIG. 12 shows a three-way array (woofers 111, 112, midranges 113, 114, and tweeters 115, 116, 117, 118) as part of a television receiver. Also indicated in this figure is a variation on the tweeter configuration to control vertical dispersion in order to reduce reflections, a configuration sometimes used in home theater equipment. Of course, other vertical arraying methods may be used

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as well. FIGs. 11 and 12 indicate that the placement of
the various acoustical emitters may deviate from a
straight line as seen by the listener; although not
normally an optimal arrangement, since it imposes some
5   path-length variations of its own as the listener
adjusts his or her seating height, some amount of such
variation is acceptable and may be an acceptable trade-
off in light of other design goals such as cabinet
design or aesthetics. The reason that some such
10   variation is acceptable is that it is known that path
length differences caused by vertical displacement of
drivers (as well as vertical) reflections from the
floor, ceiling or furniture) have a much smaller effect
on horizontal imaging than do horizontal displacements
15   (or reflections) of similar magnitudes.

What has been described is an invention which uses
a horizontally diverse array of audio emitters operating
over various audio frequency bands, corresponding
crossovers and time alignment circuits as needed, and
20   imaging circuits adapted for the particular geometry of
the audio emitters and associated crossovers, so that an
enlarged sweet spot may be enjoyed by one or more
listeners.

While specific embodiments of the audio
25   reproduction system according to the invention have been
described for the purpose of illustrating the manner in
which the invention may be made and used, it should be
understood that implementation of other variations and
modifications of the invention and its various aspects
30   will be apparent to those skilled in the art, and that
the invention is not limited by these specific
embodiments described. It is therefore contemplated to
cover by the present invention any and all

modifications, variations, or equivalents that fall within the true spirit and scope of the basic underlying principles disclosed and claimed herein.

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What is claimed is:

1.   An audio reproduction system for creating an audio presentation with enlarged sweet spot, comprising:

        source means for providing an audio program of at
5   least one channel of audio signal;

        imaging means for modifying the spatial characteristics of the audio program;

        a first loudspeaker system comprising an audio emitter for a generally higher frequency band of the
10·   spatially modified audio program and an audio emitter for a generally lower frequency band of the spatially modified audio program; and a second loudspeaker system with an audio emitter for a generally higher frequency band of the spatially modified audio program ·and an
15   audio emitter for a generally lower frequency band of the spatially modified audio program, and wherein the audio emitters of the first and second loudspeaker systems are arranged in a substantially horizontal array, and where the higher-frequency audio emitters of
20   the first and second loudspeaker systems are closer to one another than the lower-frequency audio emitters of the first and second loudspeaker systems;

        crossover network means for separating and routing the spatially modified audio program to the various
25   audio emitters of the first and second loudspeaker systems;

        wherein the imaging means is adapted for the spatial geometry of the first and second loudspeaker systems and the crossover network means associated
30   therewith.

2.·   The audio reproduction system of Claim 1 in which the audio emitters of the first and second loudspeaker

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systems are displaced generally in front of a listening area and in which the higher-frequency emitter of the first loudspeaker and the higher-frequency emitter of the second loudspeaker lie generally between the

5   lower-frequency emitter of the first loudspeaker and the lower-frequency emitter of the second loudspeaker.

3.   The audio reproduction system of Claim 2 in which the audio emitters of the first and second loudspeaker

10   systems are configured in a symmetrical fashion with respect to the listening area.

4.   The audio reproduction system as in claim 3 wherein the higher and lower frequency emitters of the first and

15   second loudspeakers are located on opposite sides of a centerline of the listening area.

5.   The audio reproduction system of Claim 1 in which the imaging means comprises a crosstalk canceller.

20

6.   The audio reproduction system of Claim 4 in which the crosstalk canceller is implemented as a part of other imaging components.

25   7.   The audio reproduction system of Claim 1 in which the imaging means creates a virtual source of sound.

8.   The audio reproduction system of Claim 1 in which the crossover network means allows at least one of the

30   higher-frequency emitters and at least one of the lower-frequency emitters to emit audio signals in substantially overlapping bands of frequencies.

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9.   The audio reproduction system of Claim 1 in which
the crossover network means includes as a series
component for at least one of the audio emitters of the
first and second loudspeaker systems a time delay device
5   to time-align the audio emitter with at least one of the
remaining audio emitters.

10.   The audio reproduction system of Claim 1 in which
the higher-frequency audio emitter of the first
10   loudspeaker system and the higher-frequency audio
emitter of the second loudspeaker system are retrofitted
to other audiovisual equipment by Velcro or adhesive
fasteners or the like to accommodate their close
spacing.
15

11.   The audio reproduction system of Claim 1 in which
the first loudspeaker system and the second loudspeaker
system comprise a common structure.

20   12.   An audio reproduction system comprising:
source means for providing an audio input signal;
a horizontally diverse plurality of audio emitters
operating over a plurality of different audio frequency
bands into a listening space;
25   imaging circuit means for creating a desired
spatial characteristic of the audio signal;
frequency dividing network means to separate and
distribute portions of the audio input signal to the
plurality of audio emitters;
30   wherein the imaging means is adapted to compensate
for the spatial configuration of the plurality of audio
emitters and the frequency dividing network means
associated therewith; and,

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wherein a set of high frequency emitters of the
plurality of audio emitters is located closer to a
centerline of the listening space than a set of low
frequency emitters of the plurality of audio emitters.

13. The audio reproduction system of Claim 12 in which
some of the plurality of audio frequency bands are
generally higher in frequency than other of the audio
frequency bands.

14. The audio reproduction system of Claim 13 in which
some of the plurality of audio emitters associated with
some of the higher-frequency bands of the plurality of
audio frequency bands are displaced substantially closer
to a geometric centerline of the listening space than
some of the other audio emitters associated with the
other of the plurality of audio frequency bands.

15. The audio reproduction system as in claim 14
wherein the same audio emitters associated with the
higher frequency bands located on a first side of the
geographic centerline are driven with a first channel
signal of the audio input signal while other audio
emitters on the first side of the geographic centerline
are driven with a second channel signal of the audio
input signal.

16. A method of increasing a relative size of a sweet
spot in a sound system comprising the steps of:
        disposing a set of woofers equidistant from and at
a first angle from a centerline of a listener on either
side of the listener; and

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disposing a set of tweeters equidistant from and at a second angle from the centerline of the listener, the second angle being less than the first angle.

5   17. The method as in claim 16 further comprising the step of applying an audio signal to the set of woofers and set of tweeters.

       18. The method as in claim 16 further comprising the
10   step of quantizing an audio image of a source of the audio signal using the set of woofers and set of tweeters.
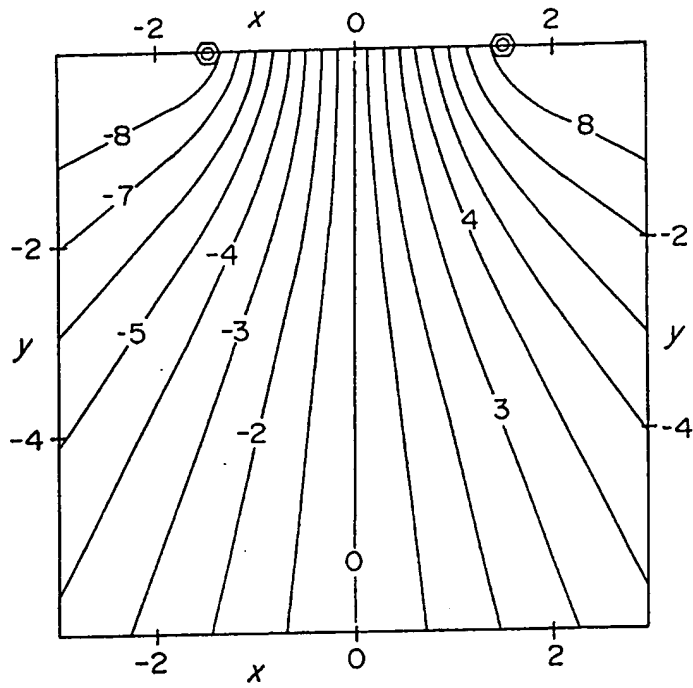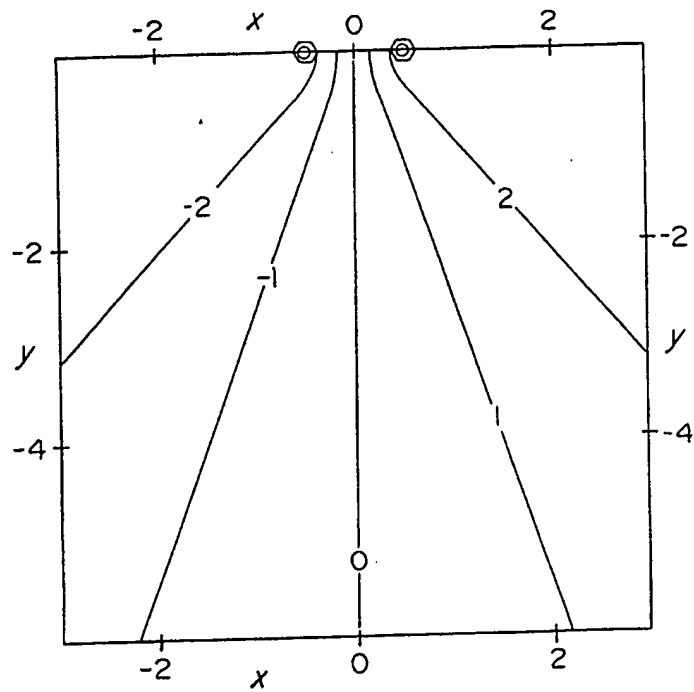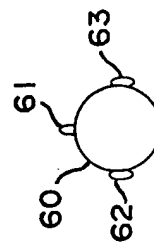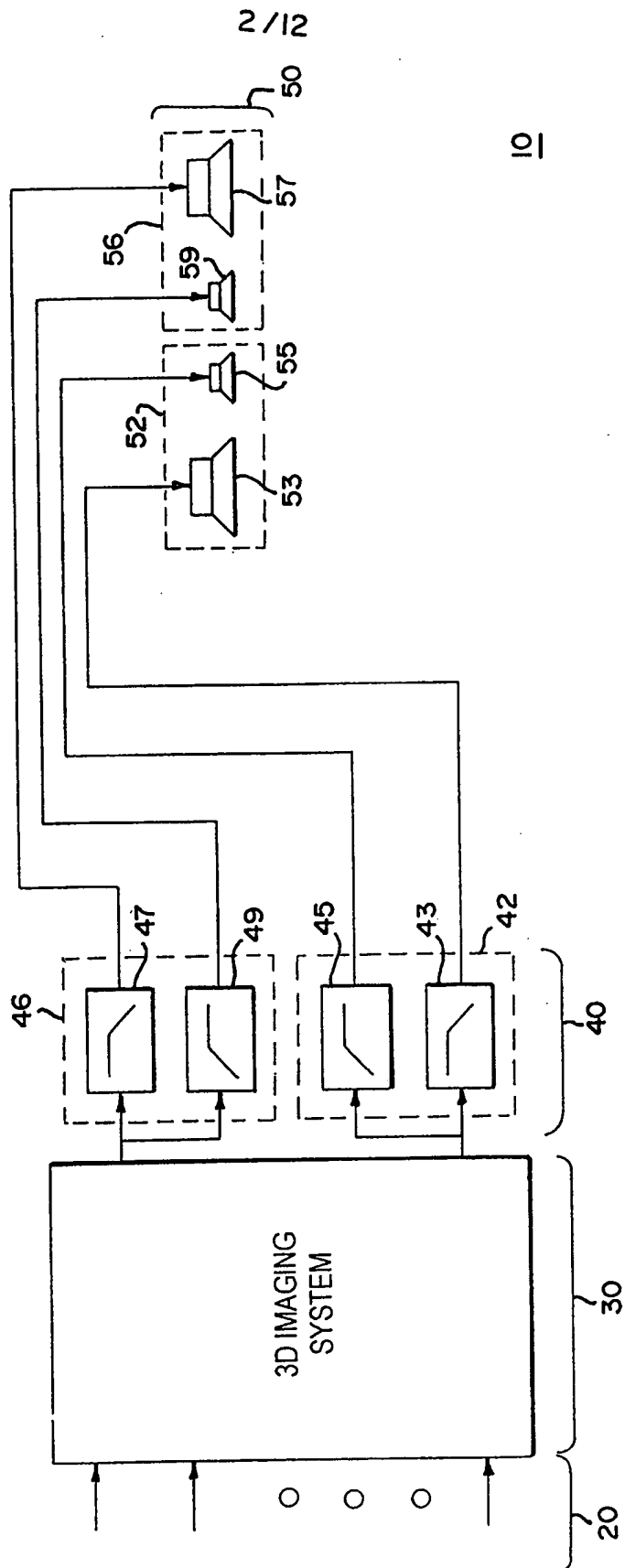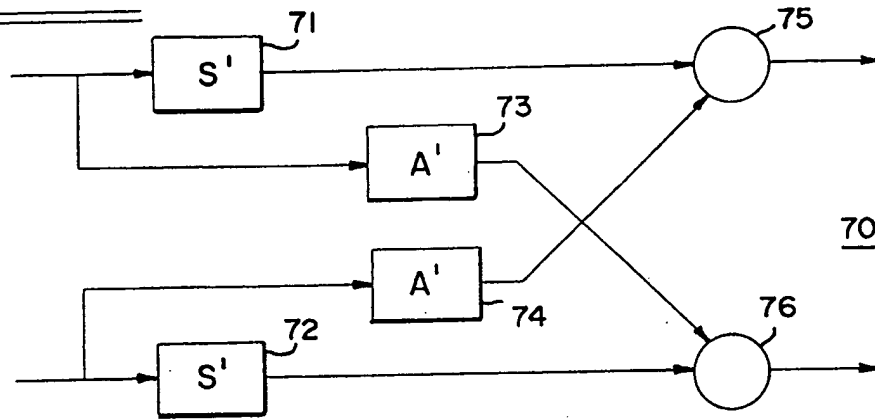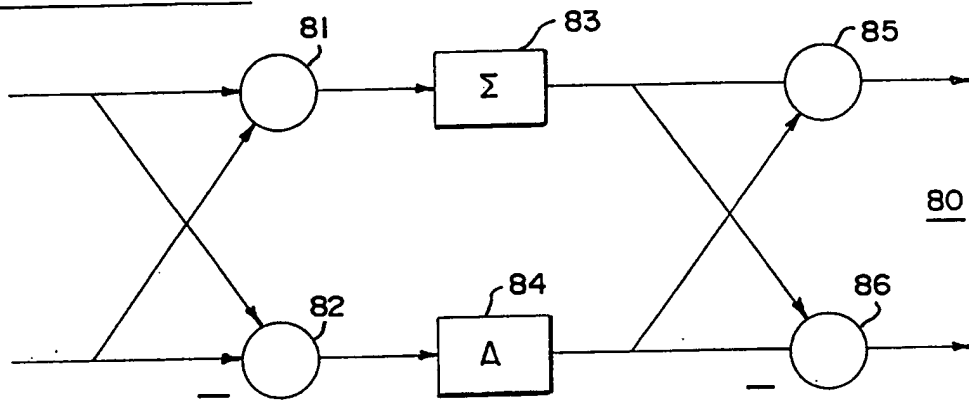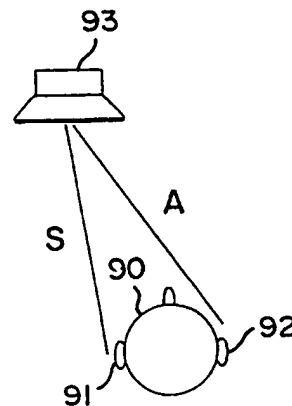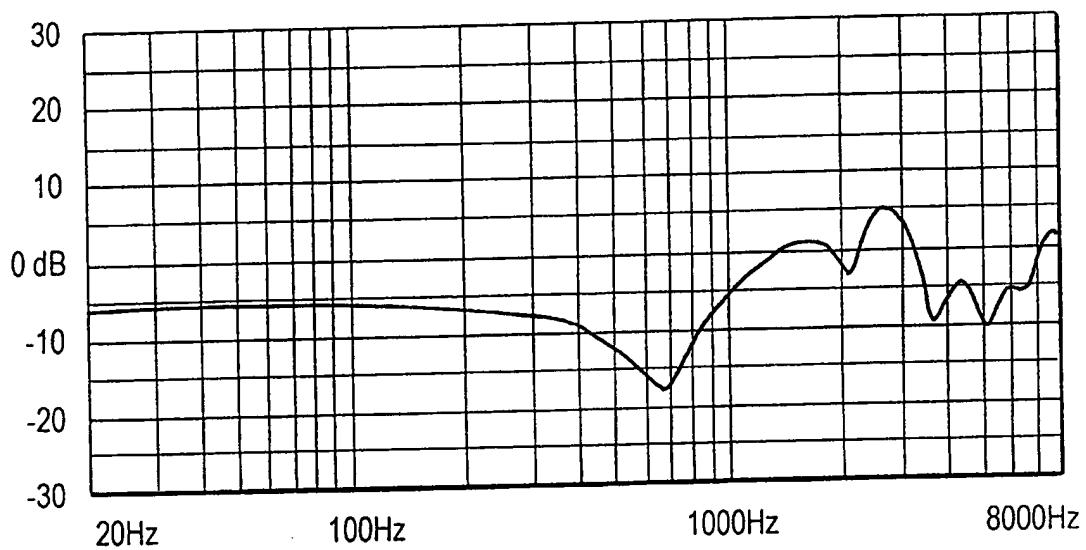
FIG. 1a

FIG. 1b

FIG. 2

FIG. 3a

70

FIG. 3b

80

FIG. 4

FIG. 5a

FIG. 5b
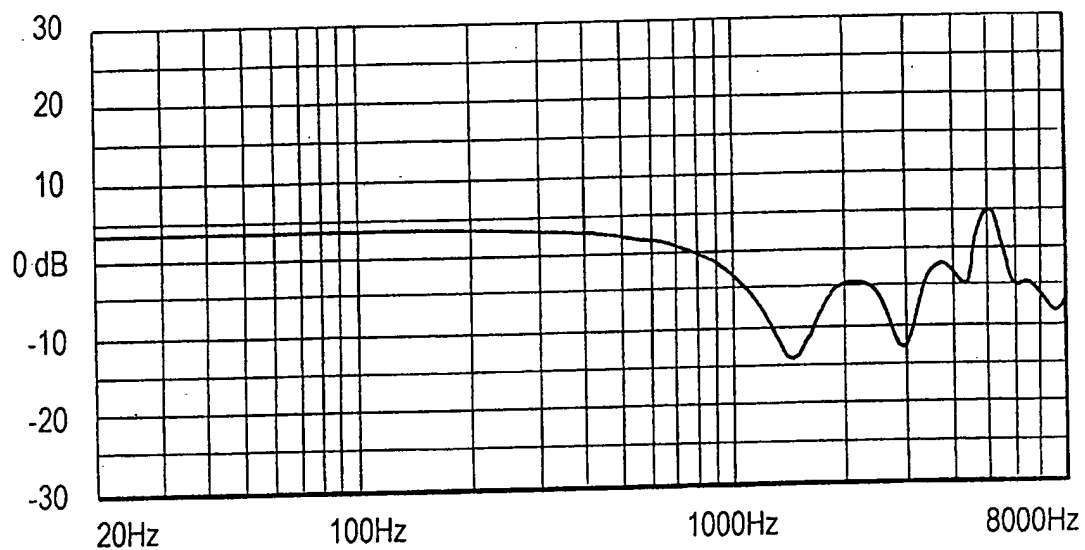
FIG. 5c
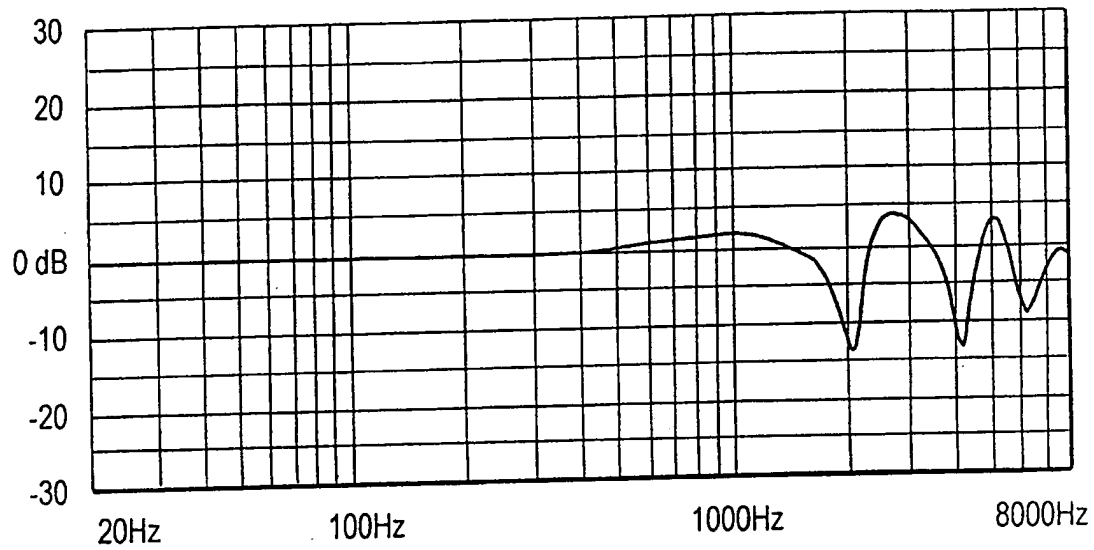
FIG. 5d
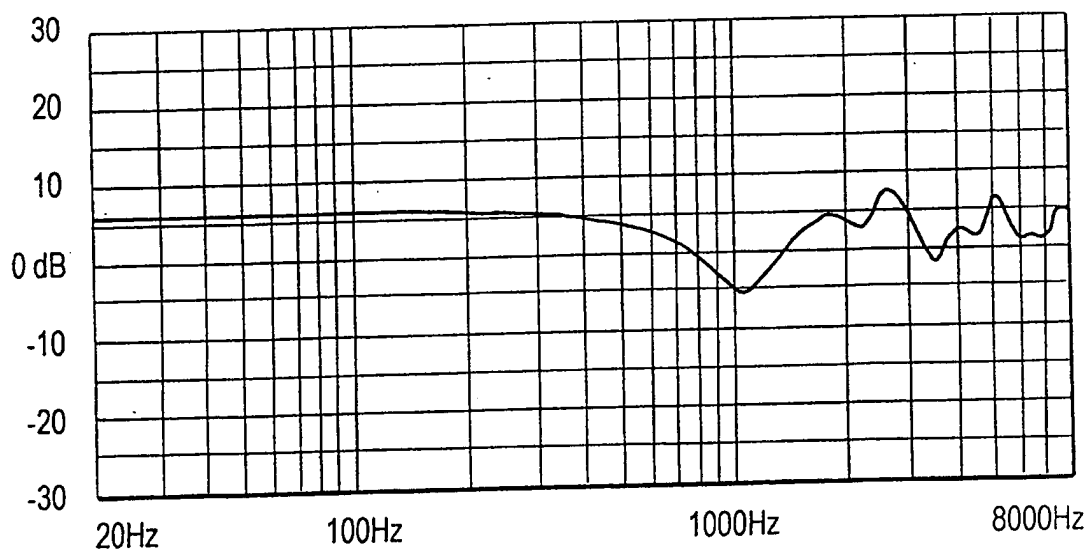
FIG. 6a
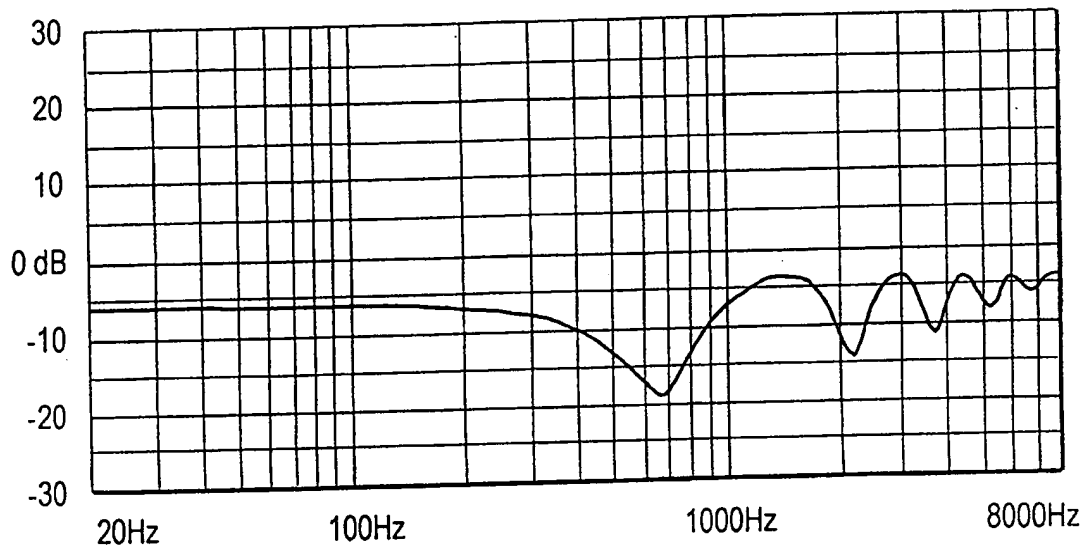
FIG. 6b
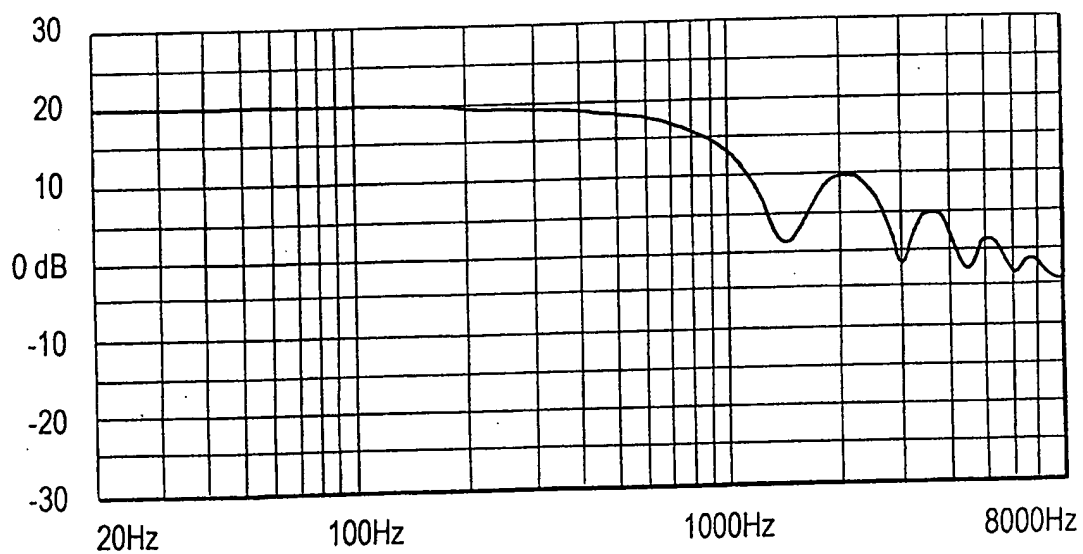
FIG. 6c
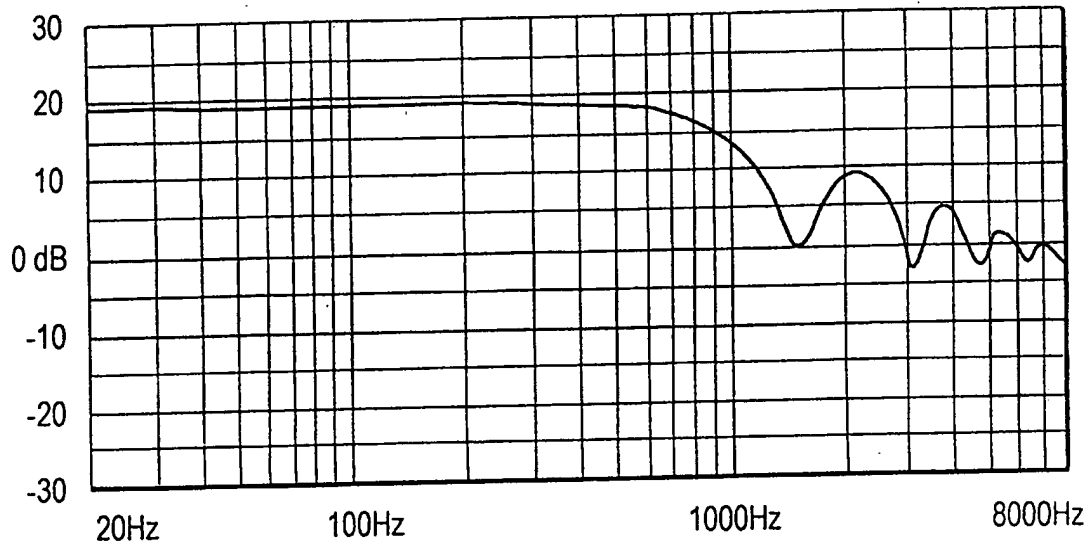
FIG. 6d
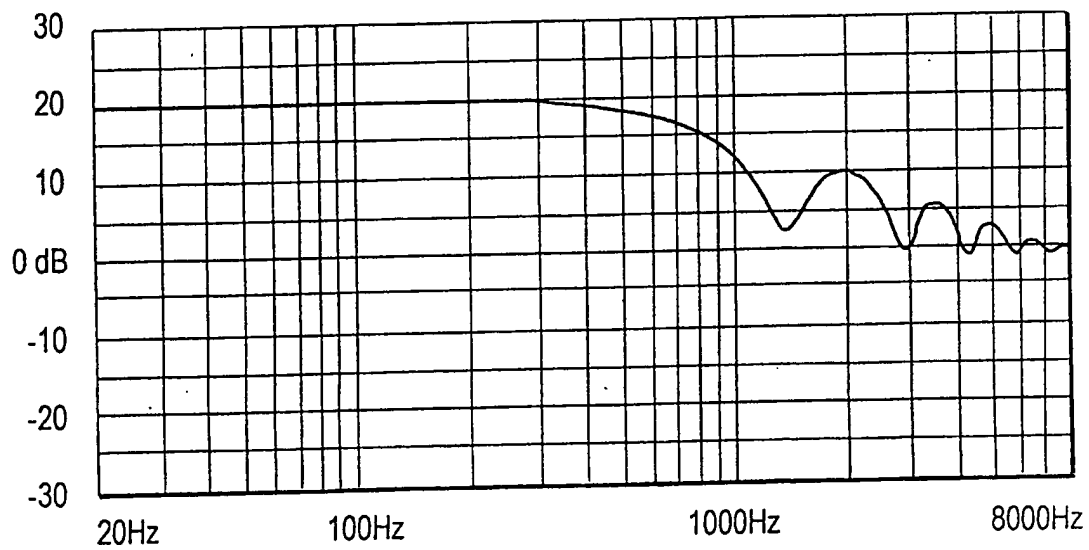
FIG. 7a
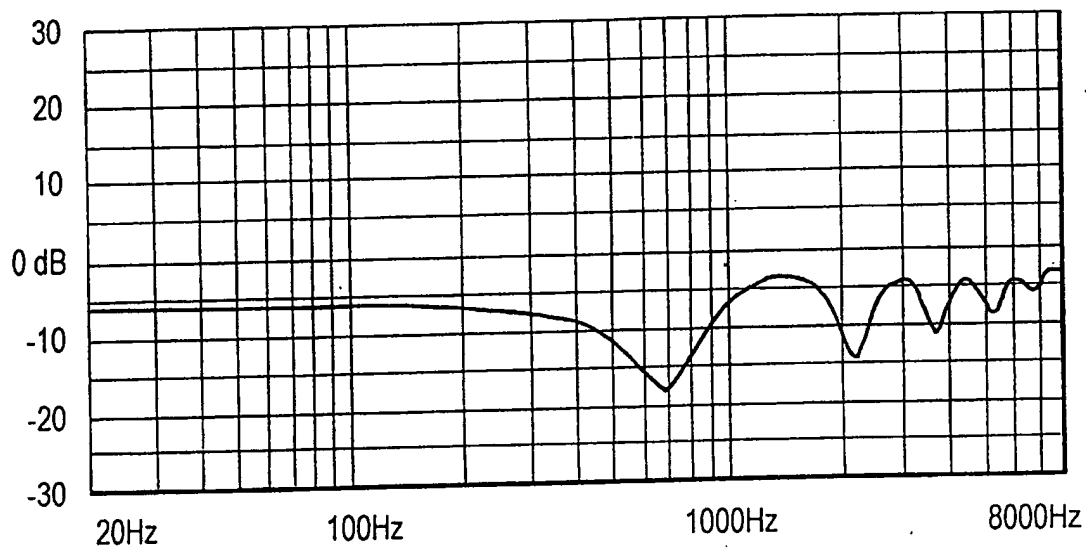
FIG. 7b
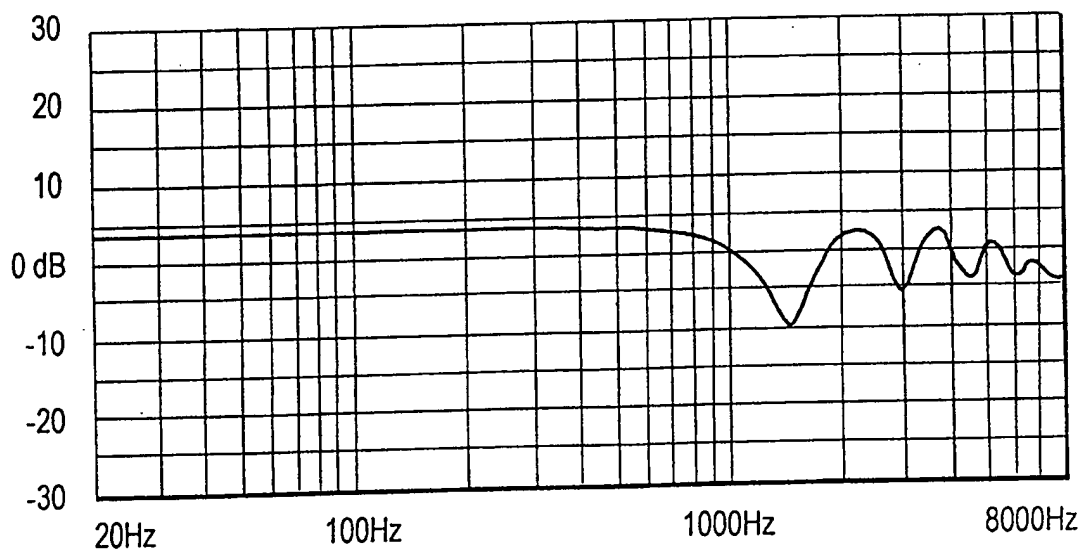
FIG. 7c
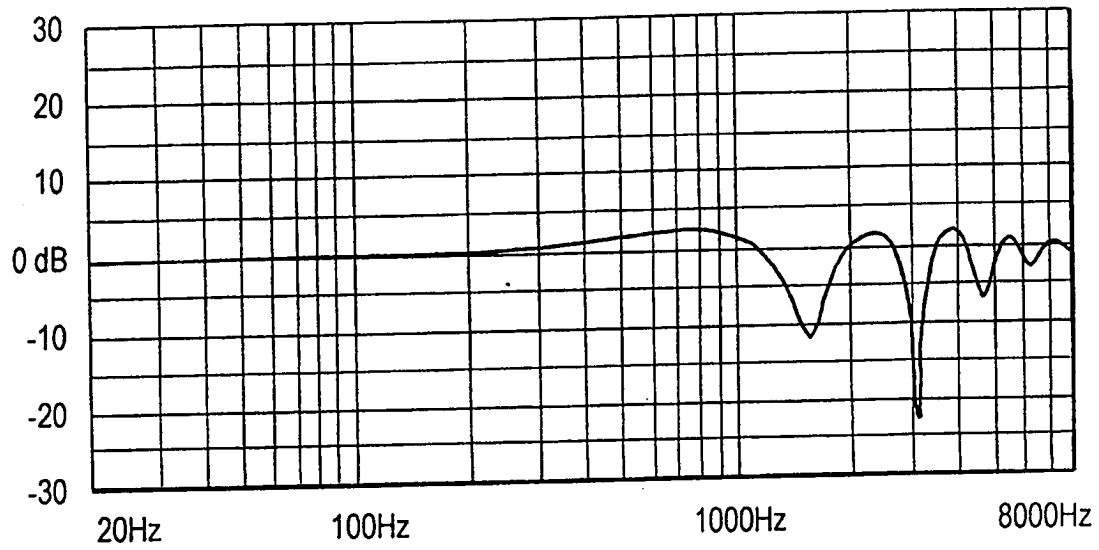
FIG. 7d
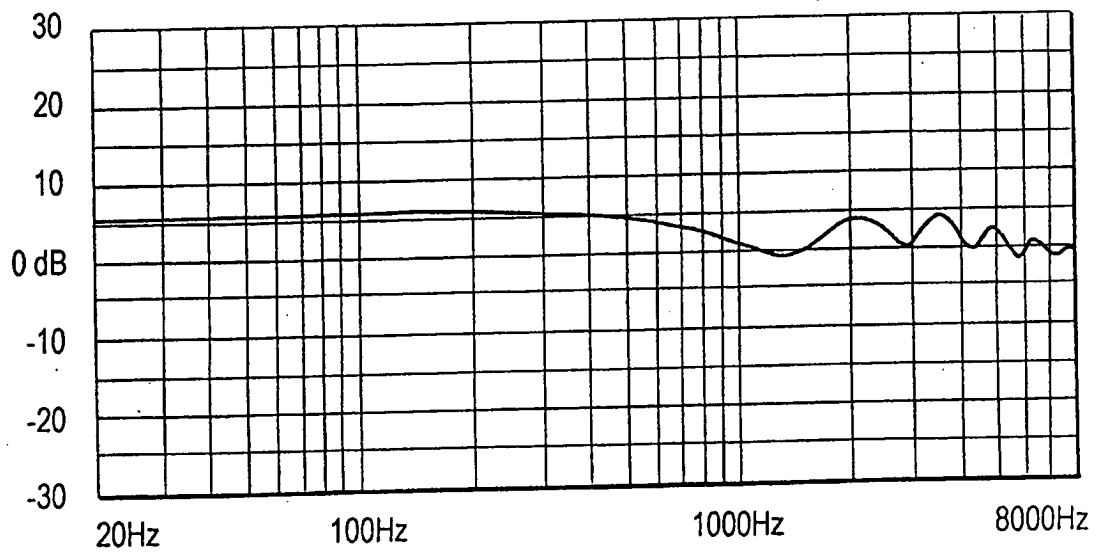
FIG. 8
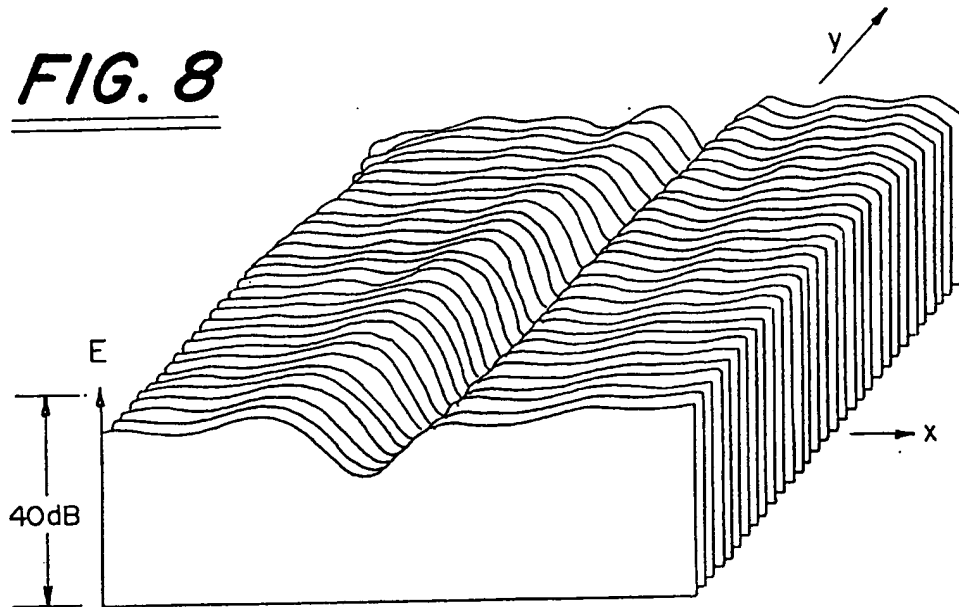
FIG. 9
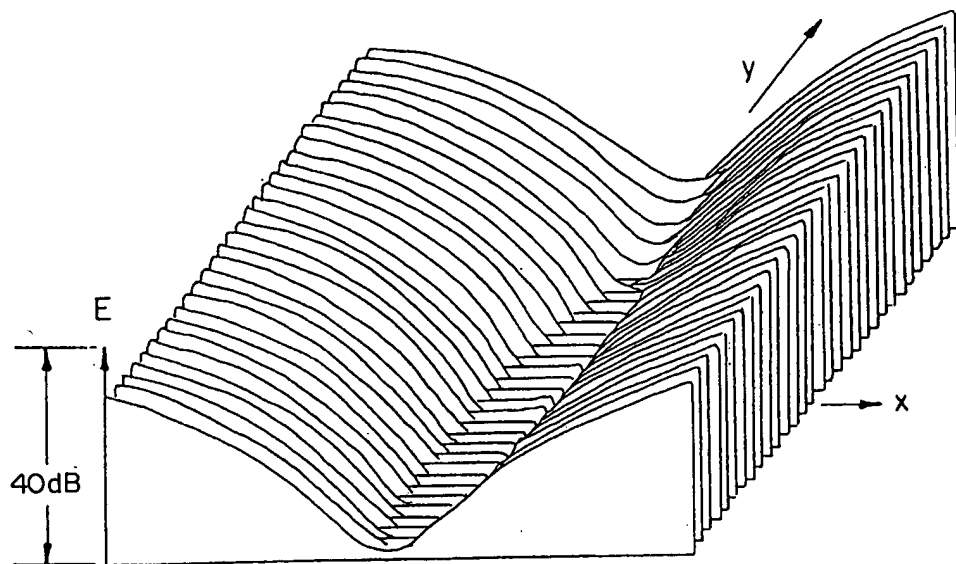
FIG. 10
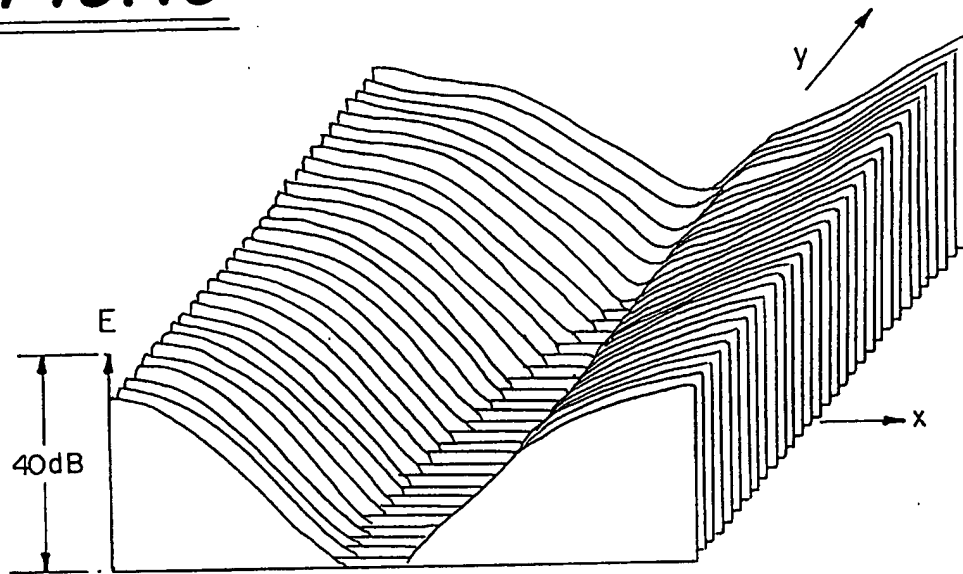
FIG. 11
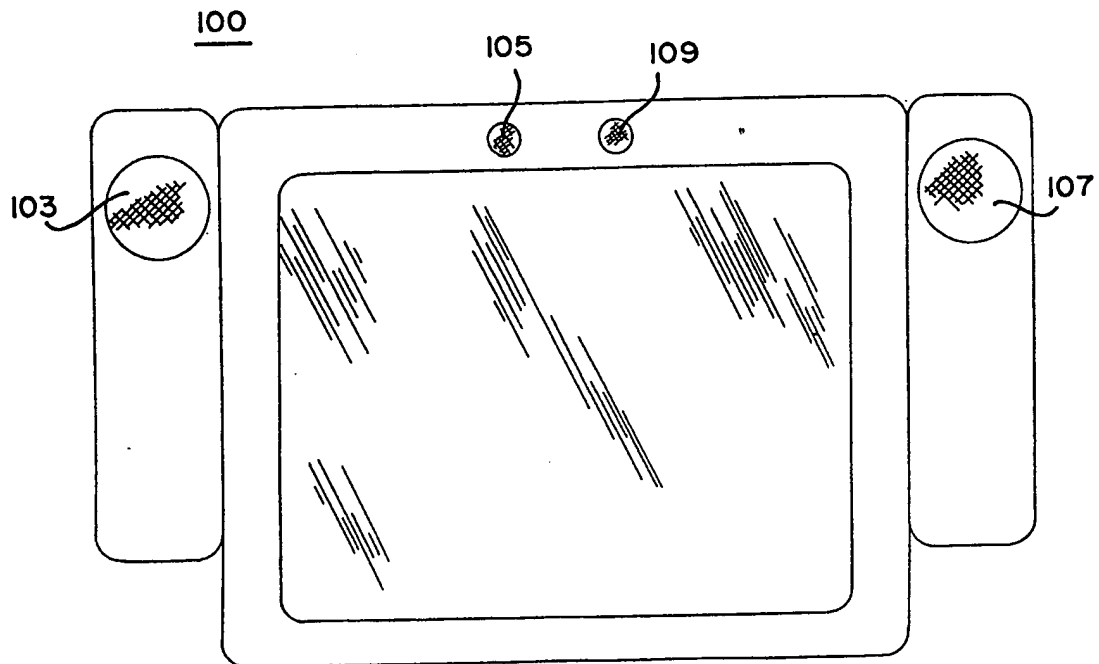
FIG. 12
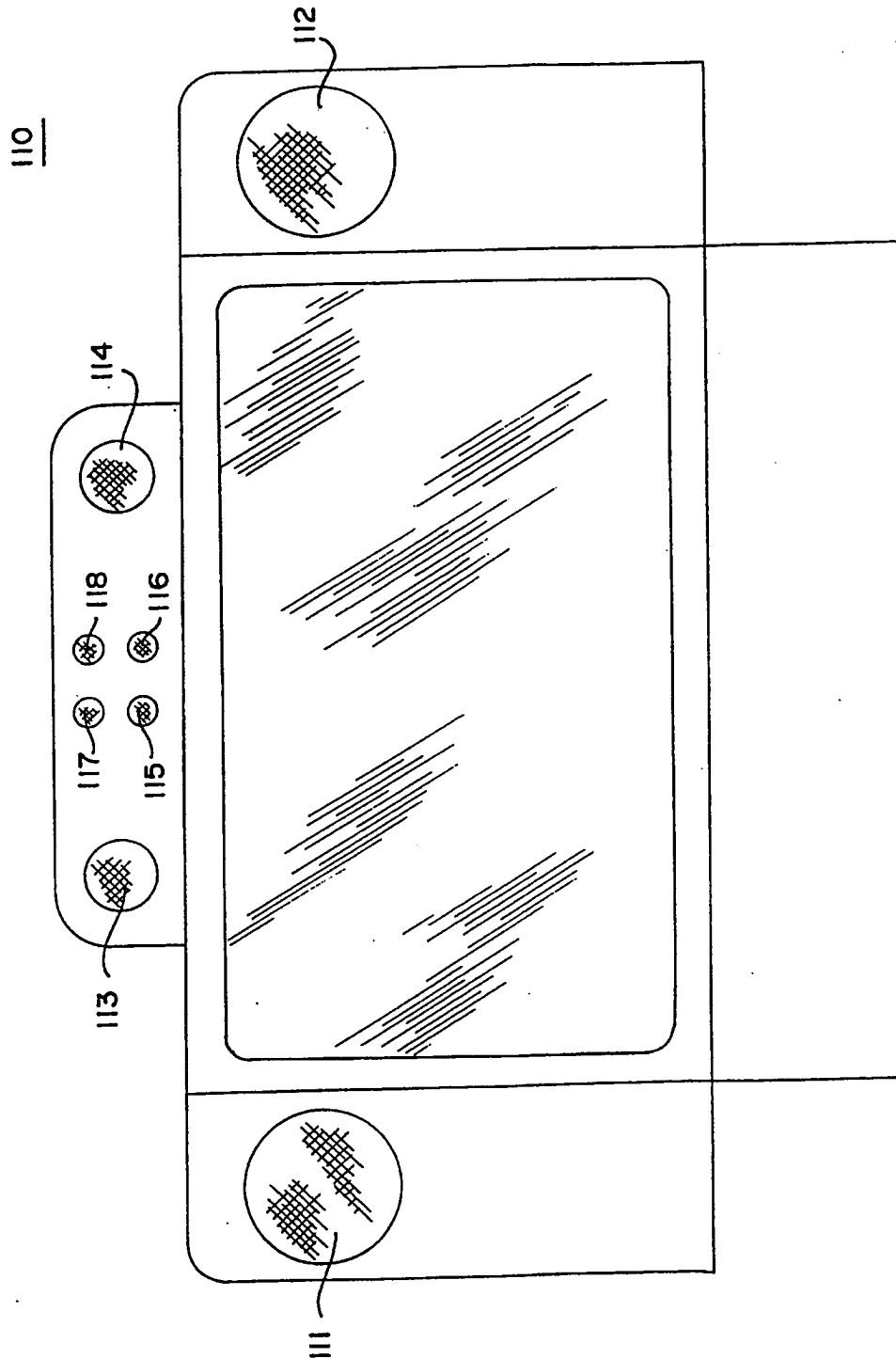
A.   CLASSIFICATION OF SUBJECT MATTER

IPC(6)   :H04R 500
US CL   :381/303, 304

According to International Patent Classification (IPC) or to both national classification and IPC

B.   FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. :   381/01, 59, 99, 300, 302-307, 332, FOR 125, FOR 165

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

None.

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS.

Search terms: sweet spot, crossover, tweeter, woofer, speakers/transducers arrangement.

C.   DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 4,961,226 A (SAFFRAN) 02 October 1990, see entire document.	1-18
X, E	US 5,764,777 A (GOLDFARB) 09 June 1998, see entire document.	1-18
X, P	US 5,751,821 A (SMITH) 12 May 1998, see entire document.	1-18
X	US 4,630,298 A (POLK et al) 16 December 1986, see entire document.	1-18
A	US 4,154,979 A (BARKER) 15 May 1979, see entire document.	1-18
A	US 4,295,006, A (TANAKA et al) 13 October 1981, see entire document.	1-18

[X] Further documents are listed in the continuation of Box C.   [ ] See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"A"	document defining the general state of the art which is not considered to be of particular relevance		
"E"	earlier document published on or after the international filing date	"X"	document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	"Y"	document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed	"&"	document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
12 AUGUST 1998	15 OCT 1998

Name and mailing address of the ISA/US	Authorized officer
Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	XU MEI
Facsimile No. (703) 305-3230	Telephone No. (703) 308-6610

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C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT		
Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 4,597,100 A (GRODINSKY et al) 24 June 1986, see entire document.	1-18
A	US 4,885,782 A (EBERBACH) 05 December 1989, see entire document.	1-18
A	US 5,297,212 A (MURAYAMA et al) 22 March 1994, see entire document.	1-18
A, P	US 5,642,429 A (JANSSEN) 24 June 1997, see entire document.	1-18
A, P	US 5,724,430 A (AARTS et al) 03 March 1998, see entire document.	1-18
A, E	US 5,778,087 A (DUNLAVY) 07 July 1998, see entire document.	1-18

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